

1. (Previously presented) A method for initiating analysis of a call from an Internet Protocol (IP) phone, the method comprising computer-implemented steps of:
 - accessing, by the phone, configuration information associated with the phone, wherein the configuration information includes information about an associated network management system;
 - automatically initiating, from the phone, transmitting an alert to the network management system, wherein the alert informs the network management system about the call and requests the analysis;
 - determining, by the phone, an issue with the call that warrants analysis of the call by performing steps selected from the group consisting of:
 - (a) determining that an elapsed time from the phone going off hook to receiving a message at the phone that instructs the phone to play a dial tone exceeds a particular value;
 - (b) transmitting to a call manager a representation of a phone number that is associated with a called party, and determining that a message was not received from the call manager in response to the representation;
 - (c) waiting for RTP packets from a called endpoint, and determining that a particular time interval has elapsed before receiving an RTP packet from the called endpoint;
 - (d) determining that a play-out time interval that is associated with a dejitter buffer that is associated with the phone is greater than a particular value;
 - (e) recording a first number of packets that are dropped before reaching the phone, by using a previous packet sequence number and a current packet sequence number, recording a second number of packets that are dropped by a dejitter buffer that is associated with the phone, by using a previous packet sequence number and a current packet sequence number, and

determining that a product of a sum of the first number of packets that are dropped before reaching the phone and the second number of packets that are dropped by the dejitter buffer, and a packetization delay that is associated with a codec that is associated with the call, is greater than a particular value;

(f) determining that RTP packets are not received continuously by the phone for a period greater than a particular value; and

(g) determining that a ratio of total packets lost before reaching the phone divided by total packets received at the phone is greater than a particular value.

2-4. (Canceled)

5. (Previously presented) The method of claim 1, wherein the particular value in the step of determining that an elapsed time from the phone going off hook to receiving a message at the phone that instructs the phone to play a dial tone exceeds a particular value is 250 ms.

6. (Canceled)

7. (Previously presented) The method of claim 1, wherein the representation of the phone number is a DTMF tone.

8. (Canceled)

9. (Previously presented) The method of claim 1, wherein the particular time interval in the step of determining that a particular time interval has elapsed before receiving an RTP packet from the called endpoint is 1000ms.

10. (Canceled)

11. (Previously presented) The method of claim 1, wherein the particular value in the step of determining that a play-out time interval that is associated with a dejitter buffer that is associated with the phone is greater than a particular value is 150 ms.
12. (Canceled)
13. (Previously presented) The method of claim 1, wherein the particular value in the step of determining that a product of a sum of the first number of packets that are dropped before reaching the phone and the second number of packets that are dropped by the dejitter buffer, and a packetization delay that is associated with a codec that is associated with the call, is greater than a particular value is 500 ms.
14. (Canceled)
15. (Previously presented) The method of claim 1, wherein the particular value in the step of determining that RTP packets are not received continuously by the phone for a period greater than a particular value is 500ms.
16. (Canceled)
17. (Previously presented) The method of claim 1, wherein the particular value in the step of determining that a ratio of total packets lost before reaching the phone divided by total packets received at the phone is greater than a particular value is 5%.
18. (Canceled)
19. (Previously presented) The method of claim 1, wherein the step of accessing includes accessing the configuration information from a remote server.
20. (Previously presented) The method of claim 19, wherein the step of accessing includes accessing the configuration information from a Trivial File Transfer Protocol server.

21-24. (Canceled)

25. (Previously presented) A computer-readable storage medium storing one or more sequences of instructions for initiating analysis of a call from an Internet Protocol (IP) phone, which instructions, when executed by one or more processors, cause the one or more processors to carry out the steps of:
- accessing, by the phone, configuration information associated with the phone, wherein the configuration information includes information about an associated network management system; and
- automatically initiating, from the phone, transmitting an alert to the network management system, wherein the alert informs the network management system about the call and requests the analysis;
- determining, by the phone, an issue with the call that warrants analysis of the call by performing steps selected from the group consisting of:
- (a) determining that an elapsed time from the phone going off hook to receiving a message at the phone that instructs the phone to play a dial tone exceeds a particular value;
 - (b) transmitting to a call manager a representation of a phone number that is associated with a called party, and determining that a message was not received from the call manager in response to the representation;
 - (c) waiting for RTP packets from a called endpoint, and determining that a particular time interval has elapsed before receiving an RTP packet from the called endpoint;
 - (d) determining that a play-out time interval that is associated with a dejitter buffer that is associated with the phone is greater than a particular value;
 - (e) recording a first number of packets that are dropped before reaching the phone, by using a previous packet sequence number and a current packet

sequence number, recording a second number of packets that are dropped by a dejitter buffer that is associated with the phone, by using a previous packet sequence number and a current packet sequence number, and determining that a product of a sum of the first number of packets that are dropped before reaching the phone and the second number of packets that are dropped by the dejitter buffer, and a packetization delay that is associated with a codec that is associated with the call, is greater than a particular value;

(f) determining that RTP packets are not received continuously by the phone for a period greater than a particular value; and

(g) determining that a ratio of total packets lost before reaching the phone divided by total packets received at the phone is greater than a particular value.

26-28. (Canceled)

29. (Previously presented) The computer-readable storage medium of claim 25, wherein the particular value in the step of determining that an elapsed time from the phone going off hook to receiving a message at the phone that instructs the phone to play a dial tone exceeds a particular value is 250 ms.

30. (Canceled)

31. (Previously presented) The computer-readable storage medium of claim 25, wherein the representation of the phone number is a DTMF tone.

32. (Canceled)

33. (Previously presented) The computer-readable storage medium of claim 25, wherein the particular time interval in the step of determining that a particular time interval has elapsed before receiving an RTP packet from the called endpoint is 1000ms.

- 34. (Canceled)
- 35. (Previously presented) The computer-readable storage medium of claim 25, wherein the particular value in the step of determining that a play-out time interval that is associated with a dejitter buffer that is associated with the phone is greater than a particular value is 150 ms.
- 36. (Canceled)
- 37. (Previously presented) The computer-readable storage medium of claim 25, wherein the particular value in the step of determining that a product of a sum of the first number of packets that are dropped before reaching the phone and the second number of packets that are dropped by the dejitter buffer, and a packetization delay that is associated with a codec that is associated with the call, is greater than a particular value is 500 ms.
- 38. (Canceled)
- 39. (Previously presented) The computer-readable storage medium of claim 25, wherein the particular value in the step of determining that RTP packets are not received continuously by the phone for a period greater than a particular value is 500ms.
- 40. (Canceled)
- 41. (Previously presented) The computer-readable storage medium of claim 25, wherein the particular value in the step of determining that a ratio of total packets lost before reaching the phone divided by total packets received at the phone is greater than a particular value is 5%.
- 42. (Canceled)

43. (Previously presented) The computer-readable storage medium of claim 25, wherein the step of accessing includes accessing the configuration information from a remote server.
44. (Previously presented) The computer-readable storage medium of claim 43, wherein the step of accessing includes accessing the configuration information from a Trivial File Transfer Protocol server.
- 45.-48. (Canceled)
49. (Previously presented) A system for initiating analysis of a call from an Internet Protocol (IP) phone, the system comprising:
- means for accessing, by the phone, configuration information associated with the phone, wherein the configuration information includes information about an associated network management system; and
 - means for automatically initiating, from the phone, transmitting an alert to the network management system, wherein the alert informs the network management system about the call and requests the analysis;
 - means for determining, by the phone, an issue with the call that warrants analysis of the call by performing functions selected from the group consisting of:
 - (a) determining that an elapsed time from the phone going off hook to receiving a message at the phone that instructs the phone to play a dial tone exceeds a particular value;
 - (b) transmitting to a call manager a representation of a phone number that is associated with a called party, and determining that a message was not received from the call manager in response to the representation;
 - (c) waiting for RTP packets from a called endpoint, and determining that a particular time interval has elapsed before receiving an RTP packet from the called endpoint;

- (d) determining that a play-out time interval that is associated with a dejitter buffer that is associated with the phone is greater than a particular value;
- (e) recording a first number of packets that are dropped before reaching the phone, by using a previous packet sequence number and a current packet sequence number, recording a second number of packets that are dropped by a dejitter buffer that is associated with the phone, by using a previous packet sequence number and a current packet sequence number, and determining that a product of a sum of the first number of packets that are dropped before reaching the phone and the second number of packets that are dropped by the dejitter buffer, and a packetization delay that is associated with a codec that is associated with the call, is greater than a particular value;
- (f) determining that RTP packets are not received continuously by the phone for a period greater than a particular value; and
- (g) determining that a ratio of total packets lost before reaching the phone divided by total packets received at the phone is greater than a particular value.

50-51. (Canceled)

52. ~~(Previously presented)~~ Currently amended A system for initiating analysis of a call from an Internet Protocol (IP) phone ~~An Internet Protocol (IP) telephony system that can initiate analysis of a call~~, the system comprising:
- a network interface;
 - a processor coupled to the network interface and receiving messages from a network through the network interface;
 - a computer-readable medium comprising one or more stored sequences of instructions which, when executed by the processor, cause the processor to carry out the steps of:

accessing, by the phone, configuration information associated with the phone, wherein the configuration information includes information about an associated network management system; and

automatically initiating, from the phone, transmitting an alert to the network management system, wherein the alert informs the network management system about the call and functions as a request for the analysis;

determining, by the phone, an issue with the call that warrants analysis of the call by performing steps selected from the group consisting of:

- (a) determining that an elapsed time from the phone going off hook to receiving a message at the phone that instructs the phone to play a dial tone exceeds a particular value;
- (b) transmitting to a call manager a representation of a phone number that is associated with a called party, and determining that a message was not received from the call manager in response to the representation;
- (c) waiting for RTP packets from a called endpoint, and determining that a particular time interval has elapsed before receiving an RTP packet from the called endpoint;
- (d) determining that a play-out time interval that is associated with a dejitter buffer that is associated with the phone is greater than a particular value;
- (e) recording a first number of packets that are dropped before reaching the phone, by using a previous packet sequence number and a current packet sequence number, recording a second number of packets that are dropped by a dejitter buffer that is associated with the phone, by using a previous packet sequence number and a current packet sequence number, and determining that a product of a sum of the first number of packets that are dropped before reaching the phone and the second number of packets that are dropped by the dejitter buffer, and a packetization delay that is

associated with a codec that is associated with the call, is greater than a particular value;

(f) determining that RTP packets are not received continuously by the phone for a period greater than a particular value; and

(g) determining that a ratio of total packets lost before reaching the phone divided by total packets received at the phone is greater than a particular value.

53. (New) The system of Claim 49, wherein the particular value in the means for determining that an elapsed time from the phone going off hook to receiving a message at the phone that instructs the phone to play a dial tone exceeds a particular value is 250 ms.
54. (New) The method of claim 49, wherein the representation of the phone number is a DTMF tone.
55. (New) The system of Claim 49, wherein the particular time interval in the means for determining that a particular time interval has elapsed before receiving an RTP packet from the called endpoint is 1000ms.
56. (New) The system of Claim 49, wherein the particular value in the means for determining that a play-out time interval that is associated with a dejitter buffer that is associated with the phone is greater than a particular value is 150 ms.
57. (New) The system of Claim 49, wherein the particular value in the means for determining that a product of a sum of the first number of packets that are dropped before reaching the phone and the second number of packets that are dropped by the dejitter buffer, and a packetization delay that is associated with a codec that is associated with the call, is greater than a particular value is 500 ms.

58. (New) The system of Claim 49, wherein the particular value in the means for determining that RTP packets are not received continuously by the phone for a period greater than a particular value is 500ms.
59. (New) The system of Claim 49, wherein the particular value in the means for determining that a ratio of total packets lost before reaching the phone divided by total packets received at the phone is greater than a particular value is 5%.
60. (New) The system of Claim 49, wherein the means for accessing includes means for accessing the configuration information from a remote server.
61. (New) The system of Claim 60, wherein the means for accessing includes means for accessing the configuration information from a Trivial File Transfer Protocol server.
62. (New) The system of Claim 52, wherein the particular value in the step of determining that an elapsed time from the phone going off hook to receiving a message at the phone that instructs the phone to play a dial tone exceeds a particular value is 250 ms.
63. (New) The method of claim 52, wherein the representation of the phone number is a DTMF tone.
64. (New) The system of Claim 52, wherein the particular time interval in the step of determining that a particular time interval has elapsed before receiving an RTP packet from the called endpoint is 1000ms.
65. (New) The system of Claim 52, wherein the particular value in the step of determining that a play-out time interval that is associated with a dejitter buffer that is associated with the phone is greater than a particular value is 150 ms.
66. (New) The system of Claim 52, wherein the particular value in the step of determining that a product of a sum of the first number of packets that are dropped before reaching the

phone and the second number of packets that are dropped by the dejitter buffer, and a packetization delay that is associated with a codec that is associated with the call, is greater than a particular value is 500 ms.

67. (New) The system of Claim 52, wherein the particular value in the step of determining that RTP packets are not received continuously by the phone for a period greater than a particular value is 500ms.
68. (New) The system of Claim 52, wherein the particular value in the step of determining that a ratio of total packets lost before reaching the phone divided by total packets received at the phone is greater than a particular value is 5%.
70. (New) The system of Claim 52, wherein the step of accessing includes accessing the configuration information from a remote server.
71. (New) The system of Claim 70, wherein the step of accessing includes accessing the configuration information from a Trivial File Transfer Protocol server.